REMARKS

In the Office Action dated April 14, 2004, claims 20, 28, 29, and 36 were rejected under 35 U.S.C. § 112, ¶ 2; claims 1, 2, 5, 29, 32-34, 39, 43, and 45 were rejected under 35 U.S.C. § 102 over Culpepper, "SIP INFO Method for Event Reporting," draft-culpepper-sip-info-event-00.txt (April 2000); claims 9-12, 18-21, 24, 25, 28, 35, 36, and 40 were rejected under § 103 over Culpepper in view of Choudhuri, "SIP INFO Method for DTMF Digit Transport and Collection," draft-choudhuri-sip-info-digit-00.txt (April 2000); claims 13 and 23 were rejected under § 103 over Culpepper in view of Choudhuri and Media Gateway Control Protocol (MGCP), Version 1.0 (hereinafter "MGCP"); claims 3 and 30 were rejected under § 103 over Culpepper in view of MGCP; claims 4, 6-8, 14-17, 22, 26, and 27 were rejected under § 103 over Culpepper in view of Choudhuri, MGCP, and Bearer Independent Call Protocol (BICP) ITU Recommendation Q.1901; claims 31, 37, and 44 were rejected under § 103 over Culpepper in view of BICP; claim 38 was rejected under § 103 over Culpepper in view of Donovan, "The SIP INFO Method," dated October 1999; and claims 41 and 42 were rejected under § 103 over Culpepper in view of Choudhuri and Donovan.

As discussed in Applicant's previous Reply to Office Action, which is incorporated by reference here, Culpepper describes the use of the SIP INFO method for communicating *mid-call* events in SIP sessions or for carrying *mid-session* signaling messages. Culpepper at 1. As a result of this teaching of Culpepper, Applicant explained that Culpepper relates to using a SIP INFO message to communicate information *after* a bearer path over a network has been established.

In response to these arguments, the present Office Action made two points. First, the present Office Action stated that the term "call" can be construed broadly so that "mid-call" as described in Culpepper can be read upon the claimed element of receiving information *before* establishing a bearer path over a network. Such a construction is unreasonable in view of the objective evidence that existed at the time of the present invention, and based on explicit expressions of Culpepper itself. The explicit statement

of Culpepper is that the SIP INFO message is used for communicating mid-call events *in SIP sessions*. Culpepper at 1. Going to the SIP Specification (RFC 2543), cited in Applicant's Information Disclosure Statement dated December 29, 2000, the term "session" is defined as a multimedia session that is a set of multimedia senders and receivers *and the data streams flowing from senders to receivers*." RFC 2543, at 10. The term "call" is used interchangeably by the SIP specification with the term "session." *See* RFC 2543, at 6 ("The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify, and terminate multimedia *sessions or calls*."). Thus, the SIP Specification provides objective evidence that a person of ordinary skill in the art at the time of the present invention would understand that the terms "session" or "call" as used in the SIP context, which is the context of Culpepper, refer to a connection in which a bearer path ("data streams flowing from senders to receivers") has been established. Therefore, "mid-call" or "mid-session" as used in Culpepper must necessarily mean *after* a bearer path has been set up.

Culpepper itself also cites to reference [4], E. Zimmerer et al., "SIP Best Current Practice for Telephony Interworking," IETF, October 1999 (copy attached). Reference [4] (Zimmerer) cited by Culpepper states that the "intent of the INFO method is to allow for SIP to carry session related control information that is generated *during a session*. The INFO method is added specifically to allow PSTN signaling messages beyond call setup and teardown to be transmitted between MGCs." Zimmerer at 6.

This statement in Zimmerer is consistent with the newly cited reference of the Office Action: Donovan, "The SIP INFO method," draft-ietf-sip-info-method-00.txt (October 1999). Donovan also states that the INFO method "is to allow for the carrying of session related control information that is generated *during a session*." Donovan at 1. As explained by Donovan, "there is no general-purpose mechanism to carry session control information along the SIP signaling path *during the session*." *Id.*

Thus, based on the objective evidence, it is clear that Culpepper contemplates the communication of information *after a bearer path has been established* (mid-call or mid-session).

The Office Action further made the second point that a person of ordinary skill in the art "must understand that in the voice over IP system, all of the signaling between two

end users is transmitted over a signaling system before establishing a connection path through data packet network." 4/14/2004 Office Action at 12. This statement, even if true, does not lead to the conclusion in the Office Action that Culpepper discloses "collecting DTMF digits by using SIP protocol in order to setup a call." *Id.* In the SIP context, the call request to establish a call session is the INVITE message. The INVITE message contains an identifier of the destination such as the INVITE message shown on page 122 of the RFC 2543 (the SIP Specification). In the example on page 122 of the SIP Specification, the SIP INVITE message contains the following address: sip: watson@boston.bell-tel.com. In the SIP context, the call request (INVITE) does not rely upon collection of DTMF digits, but rather contains a logical identifier of the destination. The statement in the Office Action (page 12) that DTMF digits are collected by the SIP protocol to set up a call is clearly erroneous--no such collection of DTMF digits is possible using the INVITE message, which is the message for establishing a call session in the SIP context. In fact, collection of DTMF digits for establishing a call is *completely unnecessary*, as the INVITE message contains a logical identifier that is sufficient for establishing the desired call between two endpoints. The DTMF digit collection referred to by Culpepper is a mid-session event, i.e., an event that transpires after the bearer path has been established between endpoints for the session. Thus, a person of ordinary skill in the art would *not* understand or recognize that SIP employs DTMF digits to establish a call.

The Office Action further cited to page 6 (Section 6) of Culpepper of teaching the collection of digits "for establishing a call connection." 4/14/2004 Office Action at 3. Section 6 of Culpepper describes examples of a SIP INFO method that can be used to request DTMF digits be detected and reported. However, the SIP INFO method referred to in Section 6 is the SIP INFO method communicated in *mid*-call or *mid*-session. As described in section 2 of Culpepper, the SIP INFO method is for "carrying DTMF digits generated *during a session*." This clearly does not satisfy the claim element "receiving the information before establishing a bearer path."

Therefore, claim 1 is clearly not anticipated by Culpepper. Independent claim 29 is similarly allowable over Culpepper.

Claims dependent from independent claims 1 and 29, including newly added dependent claim 46, are allowable over the cited references for at least the same reasons as corresponding independent claims. Claim 39, which depends from claim 1, recites that receiving the information comprises receiving the information in a Session Initiation Protocol Info message prior to establishing a bearer path over the network in response to the call request. Claim 39 recites subject matter that is *contradicted* by the teachings of Culpepper, which relates to receiving information in SIP INFO messages communicated along a SIP signaling path in mid-session or mid-call (i.e., after establishment of a bearer path).

Claim 34, which depends from claim 29, is also allowable over Culpepper. Claim 34 recites the establishing of a bearer path over the packet-based network *after receiving the information*—in contrast, in Culpepper, the information is received mid-session or mid-call, and thus, the bearer path has already been established.

Moreover, claim 11 (which depends from claim 1), further recites that requesting the information comprises requesting the information in response to determining that additional digits are desired to *establish a call.* Claim 11 was rejected as being obvious over Culpepper and Choudhuri. Neither Culpepper nor Choudhuri teaches or suggests requesting information in response to determining that additional digits are desired to establish the call. Both the proposed technique of Culpepper and the mechanism described in Choudhuri relate to mid-session or mid-call communication of the SIP INFO message for transporting DTMF digits. Both Culpepper and Choudhuri assume that a call has *already* been established--therefore, neither reference would determine that additional digits are desired to establish a call, and requesting information in response to such determining. The hypothetical combination of Culpepper and Choudhuri fails to teach or suggest the subject matter of claim 1--therefore, a *prima facie* obviousness rejection has not been established with respect to claim 1.

Independent claim 12 was rejected as being obvious over the asserted combination of Culpepper and Choudhuri. Claim 12 recites an apparatus that includes a controller to receive a call request from a media gateway controller, to determine if at least one digit is required to *establish a call session*, and to receive the at least digit from

the media gateway controller over the packet-based network from the media gateway controller in response to determining that the at least one digit is required.

Note that claim 12 recites determining if a digit is required to *establish* a call session, and to receive such digit for *establishing* a call session from the media gateway controller. This implies that the determining and receiving acts are performed *prior* to establishment of a call session. As noted above, Culpepper teaches using the SIP INFO message for communicating *mid-call* events in SIP sessions. The Office Action cited Section 6 of Culpepper as teaching the collection of digits "for establishing a call connection." 4/14/2004 Office Action at 5. Section 6 makes no reference of collecting DTMF digits with SIP INFO methods for establishing a call session. As discussed above, the collection of the DTMF digits in Culpepper is performed mid-call, i.e., *after* the call connection has been established.

Choudhuri also describes using SIP INFO messages to perform *mid-session* signaling. Choudhuri at 1-2. Therefore, even if the asserted combination of Culpepper and Choudhuri is proper, such a combination does not teach or suggest determining if a digit is required to establish a call session and receiving that at least one digit from a media gateway controller in response to determining that the at least one digit is required.

The Office Action further stated that "[i]t would have been obvious to one of ordinary skill in the art at the time the invention was made to apply collection method disclosed by Choudhuri into Culpepper's system in order to control a call between a calling party and a called party over Data network by using DTMF signal." 4/14/2004 Office Action at 5. Note that claim 12 does not refer merely to the "control" of a call--claim 12 recites determining if at least one digit is required to *establish* a call session. The mid-call or mid-session communication performed by Culpepper and Choudhuri has nothing to do with *establishing* a call session. Therefore, the hypothetical combination of Culpepper and Choudhuri fails to disclose or suggest the claimed invention. A *prima facie* case of obviousness has thus not been established with respect to claim 12.

Claims dependent from independent claim 12 are allowable over the cited references for at least the same reasons as claim 12. Moreover, claim 15, which indirectly depends from claim 12, further recites that the controller is adapted to request the at least one digit from the media gateway controller over the packet-based network in

response to determining that the at least one digit is required to establish the call session. Contrary to the assertion made in the Office Action, neither Culpepper nor Choudhuri discloses a controller to request the at least one digit from the media gateway controller over a packet-based network in response to determining that the at least one digit is required to establish the call session.

The Office Action stated that "Culpepper discloses on pages 2 and 7, second paragraph that the use of a Package designator in the event request also helps reduce any ambiguity in which media source event detection and reporting is desired for. Therefore, it implies that the system adapts with a packet-based network for requesting digits from the media gateway ...." 4/14/2004 Office Action at 8. However, such a teaching has nothing to do with a controller requesting the at least one digit from the media gateway controller over the packet-based network *in response to determining that the at least one digit is required to establish the call session*, as recited in claim 15. Although claim 15 was rejected as being obvious over Culpepper, Choudhuri, MGCP, and BICP, the Office Action provided no explanation regarding how the MGCP and BICP references cure the defects of Culpepper and Choudhuri. The obviousness rejection of claim 15 fails for this further reason.

Claim 18, which depends from claim 12, recites that the controller is further adapted to *complete a call session in response to receiving the at least one digit.* The SIP INFO messages exchanged in mid-session described in Culpepper and Choudhuri cannot satisfy this element.

Claim 41, which depends indirectly from claim 12, was rejected as being obvious over Culpepper, Choudhuri, and Donovan. Claim 41 recites that the controller is adapted to receive at least one digit in a Session Initiation Protocol Info message prior to establishing the call session. This is contradicted by the teachings of Culpepper and Choudhuri. Donovan was cited by the Office Action as sending an IAM message--however, claim 41 does not recite an IAM message. Therefore, it is unclear what relevance Donovan has with respect to claim 41. The obviousness rejection of claim 41 is thus also defective.

With respect to dependent claim 42, which depends from claim 41, neither Culpepper nor Choudhuri discloses receiving a digit in a SIP Info message prior to the

controller sending a SIP OK message. This missing element is also not taught or suggested by Donovan. Therefore, the obviousness rejection of claim 42 over Culpepper, Choudhuri, and Donovan is also defective.

Independent claim 20 was also rejected over the asserted combination of Culpepper and Choudhuri. Claim 20 recites that *prior* to a call session being established in response to a call request, a controller is adapted to receive a request to collect digits from a media gateway controller over a packet-based network. As noted above, Culpepper and Choudhuri teach communicating events during (not prior to) call session establishment. Therefore, the hypothetical combination of Culpepper and Choudhuri does not teach or suggest this element. A *prima facie* case of obviousness has thus not been established with respect to claim 20.

Claims dependent from independent claim 20 are allowable over the cited references for at least the same reasons as claim 20. Moreover, claim 26, which depends from claim 20, recites that the controller is adapted to transmit the digits within a Session Initiation Protocol message prior to the call session being established. The Session Initiation Protocol Info messages used by Culpepper and Choudhuri are communicated in mid-call or mid-session, that is, after a call session has been established. Claim 26 was rejected over Culpepper, Choudhuri, and MGCP, and BICP. Claim 26 does not recite the IAM message referred to in the Office Action--therefore, it is unclear what relevance the MGCP and BICP references have with respect to claim 26.

Independent claim 37 was rejected over the hypothetical combination of Culpepper and BICP. Even if the asserted combination of Culpepper and BICP is proper, such a combination does not teach or suggest receiving at least one digit in one of a BICC and Session Initiation Protocol message from a media gateway controller *before* establishing a voice path over a packet-based network. A *prima facie* case of obviousness has thus not been established with respect to the claim.

Claims dependent from independent claim 37 are allowable over the cited references for at least the same reasons as claim 37. Moreover, with respect to claim 44, which depends from claim 37, neither Culpepper nor BICP discloses receiving a digit in a SIP Info message *prior* to establishing a call session in response to an Invite message.

In view of the foregoing, it is respectfully submitted that all claims are in condition for allowance, which action is respectfully requested. The Commissioner is authorized to charge any additional fees, including extension of time fees, and/or credit any overpayment to Deposit Account No. 20-1504 (NRT.0075US).

Respectfully submitted,

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SIP Best Current Practice for Telephony Interworking

Status of this Memo

Abstract

This document describes Inter Media Gateway Controller (MGC) communication using the Session Initiation Protocol (SIP, [1]). SIP, with certain extensions, facilitates the exchange of signaling information between an Originating MGC and a Terminating MGC to complete calls. This document describes the best current practice for using SIP to perform this function. Where possible this draft references necessary documents, and details the concepts and methods of encapsulating PSTN signaling information in SIP messages.

Note:
This document obseletes the 'SIP+: Inter-MGC Protocol'. The ideas expressed in the SIP+ document have now taken the shape of the SIP BCP-T.

1. Introduction

This document describes Inter Media Gateway Controller communication using SIP[1]. SIP can be used to communicate from a SIP end-point (Note: Here, 'SIP end-point' has to be interpreted as a non-MGC SIP User Agent) to another SIP end-point, from a SIP end-point to a Media Gateway Controller (MGC), and from one MGC to another MGC. This document details how to best use SIP to communicate from one MGC to another MGC.

This document DOES NOT describe a new protocol. The intention of this document is to detail or reference the methods, standards and tools necessary to enable MGCs to interoperate via the SIP protocol. The SIP BCP-T replaces SIP+ (which was unfortunately viewed as being a new protocol). The SIP BCP-T describes the best current practice for using SIP for Inter-MGC communication.

The SIP BCP-T facilitates the exchange of information between an Originating Media Gateway Controller and a Terminating Media Gateway Controller so that calls may be completed. When SIP is used in the MGC-to-MGC space, there are many cases where it must "bridge" PSTN networks with IP networks. To do this, SIP must be extended to transport PSTN signaling information. By extending SIP messaging, and adding PSTN signaling encapsulation functionality, the SIP BCP-T satisfies the requirements for MGC-to-MGC communication.

SIP provides the methods to set up, tear down and manage voice and data sessions. The extensions described and/or referenced in this document enable SIP to encapsulate a variety of PSTN signaling types including but not limited to SS7, and Q.931.


```
            +--------+           +--------+           +--------+
            |        |<--SIP--->| Proxy  |<--SIP--->|        |
        +---|  MGC   |           +--------+           |  MGC   |---+
        |   |        |<-----------SIP--------------->|        |   |
        |   +--------+                                +--------+   |
      SS7        |                                        |       SS7
     Signal      |                                        |      Signal
      Link   +-------+              IP              +-------+    Link
        |    |  MG   |------------ Network ------------|  MG   |     |
        |    +-------+                              +-------+     |
        |        |   \                              /    |        |
```

```
|          |    Q.931 trunk                  CAS trunk    |      |
|       SS7 trunk \                                 /  SS7 trunk |
|          |       \                              /       |      |
|       +------------------------------------------------+      |
+----|                          PSTN                        |---+
     +------------------------------------------------+
```

Figure 1: Use of the SIP BCP-T

Figure 1 shows a basic network configuration using SIP BCP-T. In this example Media Gateways are connected to the Public Switched Telephone Network (PSTN) via SS7 trunks, Q.931 trunks, and Channel Associated Signaling (CAS) trunks. The Originating Media Gateway Controller may receive a call over any of these trunks. The signaling information from these trunks must be processed by the MGC to establish the originating half of the call, and to determine the identity of the Terminating MGC required to complete the call. The originating MGC uses SIP to communicate the necessary information to the terminating MGC to complete the call. The terminating MGC must be able to establish the terminating call half on any of the supported trunk types.

The PSTN has many regional and national signaling variants which make interoperability difficult. A key design goal of the SIP BCP-T is to document a single standard method for MGCs to interoperate. Therefore, the SIP BCP-T will use international digit analysis, dialing plans, and interoperability standards from the PSTN rather than any single National or regional variants. To guard against regional variations, the SIP BCP-T is being designed for international use from the start. The basic method of routing calls will use ITU-T E.164 [2] numbering, and will adhere to international routing of telephone numbers.

The SIP BCP-T focuses on communication between MGCs, but must also address the security concerns of SIP end-points communicating with MGCs. The information contained in MGC-to-MGC messages is sensitive, and must be secured from SIP end-points accessing the network.

The SIP BCP-T reflects the design goals of efficient call setup and scalability. This document describes a SIP implementation that is intended to scale to millions of calls per hour for a world-wide network. In addition to these ambitious goals, SIP BCP-T must facilitate "bridging" PSTN networks with IP networks. To do this, SIP BCP-T must be capable of transporting PSTN signaling information. SIP BCP-T provides the methods for MGCs to set up, tear down and manage voice and data calls. The extensions detailed here allow SIP to accommodate a variety of in bound and out bound PSTN signaling types including but not limited to SS7, Q.931, and CAS.

2. SIP BCP-T Components

2.1. SIP as defined in RFC 2543.

2.2. MIME multipart

Encapsulating PSTN signaling is a major function of the SIP BCP-T. MIME multipart payloads enable SIP to carry any PSTN signaling information required. SIP BCP-T uses MIME multipart [7-11] (RFCs 2045-2049) to enable SIP messages to contain multiple payloads in the body of the message. The multipart body can consists of any combination of the following units:

SDP (Session Description Protocol) payload
ISUP payload
One or more units of any MIME type payload.

The following are the suggested encoding formats for the above-mentioned units:

a) the SDP payload (text):
Content-Type: application/SDP; charset:ISO-10646

The SDP [3] payload is plain-text. Although the default for plain-texts in MIME is US-ASCII, ISO-10646 is recommended here for the following reasons: Both SIP and SDP use ISO 10646, and the ISO 10646 character set with UTF-8 encoding can be considered a superset of the US-ASCII character set per RFC 2044 [12].

b) the ISUP payload (arbitrary binary data):
Content-Type: application/ISUP; version: one of (ETSI1, ANSI, etc)
Content-Transfer-Encoding: binary

Encapsulating SS7 and other PSTN signaling messages inside SIP BCP-T allows MGCs to be compatible with the PSTN. SIP BCP-T encapsulates and transmits the native signaling messages from one PSTN to another, essentially tunneling the PSTN signaling messages through the IP network. To this end, SIP BCP-T has been extended with application/ISUP versions for several variants of ISUP. The use of ISUP encapsulation with Content-Type "application/ISUP" allows ISUP signaling messages to be tunneled between MGCs. The use of 'version' allows differentiation between different ISUP variants. This enables the terminating MGC to recognize and parse the messages correctly, or (possibly) to reject the message if the particular ISUP variant is not supported. The idea here is to allow MGCs to specify a preference of version, so that the following scenarios are possible: "I only like application/isup; version=ETSI1" or "I accept application/isup (but don't know the details; I just pass them on to some other tool that uses them)". The tools detailed in this document allow MGCs to encpsulate any variant of ISUP.

The MGC network architecture makes possible the direct connection of the originating MGC and the terminating MGC without intermediate MGCs. This makes possible the scenario where an MGC in one country must be able to "speak" the ISUP variants of all other countries in order to complete calls, (as opposed to an intermediate international gateway MGC). Alternatively, the given MGC could use only a superset ISUP protocol, or an agreed

-upon "lowest common denominator" ISUP variant, which all other MGCs connected to that MGC would have to use. The ability to encapsulate any version of ISUP inside SIP messages enables any of these scenarios. The optimal interworking of protocol variants can be determined by the network operator. A superset approach, an agreed upon (lowest common denominator) interworking variant approach, or support of all variants approach can all be implemented using the SIP BCP-T.

For example, when a call arrives at an MG on an SS7 trunk, the Originating MGC encapsulates the IAM in the INVITE message body that is sent to the terminating MGC. This MGC reads the IAM from the INVITE payload and may use it when creating its signaling message to the terminating telephone network.

Subsequent ACM and ANM messages are passed back to the Originating MGC along with other necessary information via 100 Trying and 200 OK messages. The version tells the receiving MGC which type of ISUP message has been encoded.

Note: 100 responses aren't forwarded by intermediate servers and this constitutes a problem while trying to encapsulate the ISUP ACM. One method to get around this is to introduce a new 100 class message that would go right upto the originating MGC, and use this to carry the ACM. It should be noted that this idea is still in a nascent stage and warrants more discussion at this point of time. Efforts are on to determine whether a proposal to deal with this currently exists, or if the formulation of a new one is necessary.

c) One or more units of any MIME type payload (dependent on use):

No rules have been defined here. The Content-Type and Content-Transfer-Encoding parameters would be determined by the MIME type and the kind of data sent compliant with the rules of RFCs 2045, 2046.

Note: The Content-Type specification is mandatory in all instances to differentiate between the different payload types. This is because there is no guarantee of a specific order or required type of the payloads.

2.2.1 An illustrative example:

SIP message format requires a Request line followed by Header lines followed by a CRLF separator followed by the message body. To illustrate the use of multipart payload message body, below is an INVITE message which has the originating SDP information and an encapsulated ISUP IAM:

```
INVITE sip:13035553142@Den1.level3.com SIP/2.0
From: sip:13035553355@den3.level3.com
To: sip:13035553142@Den1.level3.com
```

```
Call-ID: DEN1231999021712095500999@Den1.level3.com
Content-Length: 377
Content-Type: multipart/mixed; boundary=unique-boundary-1
MIME-Version: 1.0

--unique-boundary-1
Content-Type: application/SDP; charset=ISO-10646
```



```
v=0
o=ezimmerer 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP seminar
c=IN IP4 MG122.level3.com
t= 2873397496   2873404696
m=audio 9092 RTP/AVP 0 3 4

--unique-boundary-1
Content-type:application/ISUP; version=ETSI1
Content-Transfer-Encoding: binary

89 8b 0e 95 1e 1e 1e 06 26 05 0d f5 01 06 10 04 00

--unique-boundary-1--
```

Note:
Since binary encoding is used for the ISUP payload, each byte is encoded as a byte, and not as a two-character hex representation. Hex digits were used in the draft because a literal encoding of those bytes would have been confusing and unreadable.

2.3. ISUP MIME Type
The ISUP MIME type is required to encapsulate the PSTN ISUP signaling information. See Internet draft: draft-zimmerer-sip-isup-mime-00.txt [4]

2.4. INFO method
The intent of the INFO method is to allow for SIP to carry session related control information that is generated during a session. The INFO method is added specifically to allow PSTN signaling messages beyond call setup and teardown to be transmitted between MGCs. This method may be used by either originating or terminating MGC to communicate additional information such as mid-call telephony signaling messages resulting from the interworking between an ISUP or ISDN network device and a SIP controlled network.

This has been described in an Internet Draft:
draft-ietf-mmusic-sip-info-method-01.txt. [5]

Note: This draft references draft-ietf-sigtran-mime-isup-00.txt proposal for encapsulating telephony signaling control information as part of an ISUP attachment to the INFO message. It is recommended to use the format outlined above in section 2.2. MIME Multipart instead.

2.5. ISUP-SIP mapping
(Internet draft pending)
The SIP BCP-T recommends the ISUP-SIP mapping detailed in the

Internet draft: draft-camarillo-mmusic-sip-isup-bcp-00.txt.
[6].

2.6. Network Access Point (NAP) architecture
(Internet draft pending)

3. Security

The security provided by SIP is sufficient for the initial deployment of inter-MGC communication. There are issues requiring further study with regard to the interoperation of MGCs and SIP end-points. There is an interesting line that must be drawn between giving too much control to the endpoint clients and not enough control. The notion of a secure proxy between SIP end-points and the network MGCs requires more study.

4. Glossary

ACM Address Complete Message. One of the ISUP call setup messages.A message sent in the backward direction indicating that all the address signals required for routing the call to the called party have been received.

ANM ANswer Message. Another one of the ISUP call setup messages. A message sent in the backward direction indicating that the call has been answered.

CAS Channel Associated Signaling. Signaling (for example, in a T-1 line) in which control signals, are carried in the same channel along with voice and data signals.

IAM Initial Address Message in the ISUP call set up messages. A mandatory message sent in the forward direction to initiate seizure of an outgoing circuit.

ISDN Integrated Services Digital Network in concept is the integration of both analog or voice data together with digital data over the same network.

ISUP ISDN User Part. This defines the methods and protocols used in the establishment and tear-down of voice and data calls over the public switched network, and to manage the trunk network on which they rely. It is used for both ISDN and non-ISDN calls.

MGC Media Gateway Controller. Also known as a Call Agent or a SoftSwitch. This is the unit that controls the media gateways and is responsible for all the control signaling necessary for call set-up.

MIME Multipurpose Internet Mail Extensions. This set of documents redefines the format of mail messages to allow for multipart message bodies, textual message

bodies and header information in character sets other than US-ASCII, etc.

PSTN Public Switched Telephone Network. It refers to the world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned.

SDP Session Description Protocol. It is a protocol that is used to transfer information between interested parties to discover and participate in a multimedia session. Fully specified as a standard in RFC 2327 ([3]).

SIP BCP-T Best Current Practices for using SIP in Telephony. It is an extension of SIP (as specified in RFC 2543) to facilitate inter-MGC communication.

SS7 Signaling System 7. It is a global standard for tele-communications that defines the architecture for performing the call-establishment, billing, routing, and information-exchange functions of the PSTN.

Q.931 This ITU-T recommendation is the signaling protocol in the PRI ISDN D channel. It describes what goes into a signaling packet and defines the message type and content.

UTF-8 Unicode Standard Transformation Format. This is a transformation format that retains the full US-ASCII range and provides compatibility with file systems, parsers and other software that rely on US-ASCII values but are transparent to other values.

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